a second output port configured to be coupled to an Internet communications network, and

a processing unit configured and arranged to analyze the audio information and, in response to the analysis, to determine [if] whether the audio information received from the telephone is to be coupled to the first output port to establish a standard telephonic communication using the standard switched telephone communications network, or if the audio information is to be processed in accordance with the standard Internet transfer protocols and coupled to the second output port to [established] establish an Internet communication using the Internet communications network to communicate the processed audio information in accordance with the standard Internet transfer protocols.

2. An arrangement as recited in claim 1, wherein the standard Internet transfer protocols include a standard gatekeeper protocol for handling gatekeeper signaling, a standard Internet call protocol for handling Internet call signaling and a standard end-to-end protocol for handling end-to-end control.

3. An arrangement as recited in claim 2, wherein the standard gatekeeper protocol uses an RAS standard protocol, the standard Internet call protocol uses a Q.931 standard protocol and the standard end-to-end protocol uses an H.245 standard control protocol.

4. An arrangement as recited in claim 1, wherein the standard Internet transfer protocols include a standard packetization protocol to packetize a stream of audio information.

5. An arrangement as recited in claim 4, wherein the standard packetization protocol uses a standard real-time transfer protocol (RTP).

6. An arrangement as recited in claim 3, wherein the standard Internet transfer protocols include a standard real-time transfer protocol (RTP) to packetize a stream of audio information.

7. An arrangement as recited in claim 4, wherein the standard Internet transfer protocols include a standard quality-of-service protocol for gathering quality-of-service statistics of packetized information delivered to a receiving device.

8. An arrangement as recited in claim 5, wherein the standard Internet transfer protocols include a standard quality-of-service protocol for gathering quality-of service statistics regarding packetized information communicated over the Internet.

9. An arrangement as recited in claim 8, wherein the standard quality-of service protocol uses standard real-time transfer control protocol (RTCP).

10. An arrangement as recited in claim 7, further comprising a monitoring unit provided to monitor the quality-of-service statistics and to adaptively control a rate at which audio information is transferred over the Internet.

11. An arrangement as recited in claim 9, further comprising a monitoring unit provided to monitor the RTCP information and to adaptively control a rate at which audio information is transferred over the Internet.

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12. (Twice Amended) A method of providing telephonic communication using an Internet communications channel, the method comprising the steps of:

providing a first communications device, coupled to a standard switched telephone network for normal telephonic communication and to an Internet connection coupled to the Internet, the first communications device including an interface device provided to selectively couple an output of the first communications device to one of the standard switched telephone network and the Internet connection, the interface device being adapted to automatically determine , in response to data information designating a communication addressee, whether [if] the output is to be selectively coupled to at least one of: the standard switched telehone network and the Internet connection;

providing a second communication device coupled to the Internet;

initiating a call using the first communication device to the second communication device using by establishing an initial Q.931 protocol;

establishing far end control of the second communication device by the first communication device in accordance with an H.245 protocol;

performing gatekeeper signaling in the first communication device accordance with an RAS protocol; and

packetizing audio information of the telephonic communication for transfer over the Internet using a standard real-time transfer protocol (RTP).

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16. (Amended) An arrangement for providing telephonic communication that may be selectively transmitted via the Internet using standard Internet protocols, comprising:

a telephone; and

interface means coupled to the telephone and configured and arranged to receive audio information [of the] designating a telephonic communication addressee, the interface means comprising:

first output means configured to be coupled to a standard switched telephone communications network,

second output means configured to be coupled to an Internet communications network, and

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processing means configured and arranged to determine [if] whether the audio information received from the telephone is to be coupled to the first output means to establish a standard telephonic communication using the standard switched telephone communications network, or [if the audio information is] to be processed in accordance with the standard Internet transfer protocols and coupled to the second output means to establish an Internet communication using the Internet communications network to communicate the processed audio information in accordance with the standard Internet transfer protocols.

17. (Amended) A method for providing telephonic communication that may be selectively transmitted via the Internet using standard Internet protocols, the method comprising:

providing an interface unit having a memory and adapted to receive telephonic communication in response to user intervention and to communicate the telephonic communication via at least one of: a first output coupled to a standard switched telephone network and a second output coupled to an Internet communications network;

providing a telephone device communicatively coupled to the interface unit;

generating audio information, that designates a communication addressee, at the telephone and sending the information to the interface unit;

analyzing the audio information and therein automatically determining, at the interface unit, [if] whether the audio information received from the telephone is to be coupled to the first or second output; and

responsive to the determination, coupling the telephone via the interface unit to at least one of the standard switched telephone network and the Internet communications network.

18. *(Amended)* The method of claim 17, wherein automatically determining [if] whether the audio information is to be coupled to the first or second output is responsive to comparing a DTMF code received as part of the audio information to a variable stored in memory at the interface and is without further user intervention.

19. *(Amended)* The method of claim 17, wherein automatically determining [if] whether the audio information is to be coupled to the first or second output is responsive to detecting a DTMF code received as part of the audio information that represents the number for a local Internet access provider and is without further user intervention.

20. (Amended) The method of claim 17, wherein automatically determining [if] whether the audio information is to be coupled to the first or second output is responsive to comparing a DTMF code received as part of the audio information to a telephone number stored in memory at the interface and is without further user intervention.

21. (Amended) The arrangement of claim 1, wherein the interface unit further comprises a memory, and wherein the processing unit is adapted to automatically determine [if] whether the audio information is to be coupled to the first or second output by comparing a DTMF code received as part of the audio information to a variable stored in memory at the interface, without further audio information.

22. (Amended) The arrangement of claim 1, wherein the processing unit is adapted to automatically determine [if] whether the audio information is to be coupled to the first or second output by detecting if a DTMF code received as part of the audio information represents the number for a local Internet access provider, without further audio information.

23. (Amended) The arrangement of claim 1, wherein the interface unit further comprises a memory, and wherein the processing unit is adapted to automatically determine [if] whether the audio information is to be coupled to the first or second output by comparing a DTMF code received as part of the audio information to a telephone number stored in memory at the interface, without further audio information.

Remarks

Favorable reconsideration of this application is requested in view of the following remarks. For the reasons set forth below, Applicant respectfully submits that the claimed invention is allowable over the cited references.

In the August 14 Office Action, the Examiner once again indicates that the "listed patent and document" from the Information Disclosure Statement ("IDS") submitted June 25, 1998 should be resubmitted. As in the previous Office Action Response, Applicant again asks for clarification on which items were lost at the USPTO from the previous submission. While this clarification has not yet been provided, Applicant submits herewith the patent documents identified in the IDS. Should other ones of the references still be needed, Applicant assumes it will be so notified.

Turning now to the rejections, claims 1, 16 and 17 stand rejected under § 102(e) as being clearly anticipated by *Kubler et al.* (U.S. Patent No. 5,726,984) and claims 1-12 and 16-23 stand